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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/302,397	04/30/1999	KAZUNORI OZAWA	SON-0432	6830

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EXAMINER	
ARMSTRONG, ANGELA A	
ART UNIT	**PAPER NUMBER**
2654	19

DATE MAILED: 08/12/2003

Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)	
	09/302,397	OZAWA, KAZUNORI	
	Examiner	Art Unit	
	Angela A. Armstrong	2654	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133).
- Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *24 May 2003* .

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-11* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-11* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

11)☐ The proposed drawing correction filed on _____ is: a)☐ approved b)☐ disapproved by the Examiner.

If approved, corrected drawings are required in reply to this Office action.

12)☐ The oath or declaration is objected to by the Examiner.

Priority under 35 U.S.C. §§ 119 and 120

13)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____ .

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

14)☐ Acknowledgment is made of a claim for domestic priority under 35 U.S.C. § 119(e) (to a provisional application).

a) ☐ The translation of the foreign language provisional application has been received.

15)☐ Acknowledgment is made of a claim for domestic priority under 35 U.S.C. §§ 120 and/or 121.

Attachment(s)

1) ☐ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO-1449) Paper No(s) _____ .
4) ☐ Interview Summary (PTO-413) Paper No(s). _____ .
5) ☐ Notice of Informal Patent Application (PTO-152)
6) ☐ Other: .

DETAILED ACTION

1. Applicant's request for reconsideration of the finality of the rejection of the last Office action is persuasive and, therefore, the finality of that action is withdrawn.

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

2. Claims 1-11 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kleijn et al (US Patent No. 5,704,003) in view of Ozawa et al, "M-LCELP Speech Coding at 4KBPS", Acoustics, Speech, and Signal Processing, 1994. ICASSP-94, 1994 IEEE International Conference on, Volume: 1, 19-22 April 1994, Page(s): I/269 -I/272 vol.1.

Regarding claims 1 and 6, Kleijn et al teaches a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter at col. 5, lines 66-67 and col. 6, lines 1-3.

Additionally, Kleijn teaches an adaptive codebook section for obtaining a delay and a gain and obtaining a residual by predicting a speech signal at col. 6, lines 52-62;

Kleijn also discloses a discrimination section for discriminating a voiced/unvoiced mode at col. 7 lines 10-26 and col. 5, lines 7-8;

Kleijn et al do not specifically teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook. However, determination of a

voiced/unvoiced mode based on a past quantized gain of an adaptive codebook was well known in the art.

In a similar field of endeavor, Ozawa et al teaches a speech coding system (implementing encoder and decoder structures) which implements mode selection such that coding methods and codebooks are changed to improve coding efficiency as well as to reduce codebook size, in accord with each of 4 modes, for unvoiced and transition segments, and voiced segments. The mode is determined by comparing the average of the pitch prediction gain with each of three thresholds (page I-269, section 2).

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the coding system of Kleijn et al to implement discriminating a voiced/unvoiced mode is based on a past quantized gain, as taught by Ozawa et al, for the purpose of improving coding efficiency and reducing codebook size, as suggested by Ozawa et al.

Kleijn et al further teach, sound source quantization which has a codebook for representing a signal by combination of pulses and amplitudes and searches code vectors stored in the codebook and delays or shift amounts so as to output a combination of code vector and shift amount that minimizes distortion at col. 6, lines 21-61.

Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction. However, implementation of a multiplexer and providing an analogous decoding scheme for a specific coding system was well known.

Ozawa implements a M-LCELP encoder and decoder structure, which includes multiplexer on the encoder and demultiplexer with the decoder.

Therefore, it would have been obvious to one of ordinary skill at the time of invention to implement a multiplexer and a decoding scheme with the system of Kleijn et al for the purpose of providing high quality speech coding and decoding as suggested by Ozawa et al.

Regarding claims 2, 5, and 7, Kleijn discloses spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter at col. 5, lines 66-67 and col. 6, lines 1-3;

Additionally, Kleijn disclose an adaptive codebook section for obtaining a delay and a gain and obtaining a residual by predicting a speech signal at col. 6, lines 52-62 and a discrimination section for discriminating a voiced/unvoiced mode at col. 7 lines 10-26 and col. 5, lines 7-8;

Kleijn et al do not specifically teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook. However, determination of a voiced/unvoiced mode based on a past quantized gain of an adaptive codebook was well known in the art.

In a similar field of endeavor, Ozawa et al teaches a speech coding system (implementing encoder and decoder structures) which implements mode selection such that coding methods and codebooks are changed to improve coding efficiency as well as to reduce codebook size, in accord with each of 4 modes, for unvoiced and transition segments, and voiced segments. The mode is determined by comparing the average of the pitch prediction gain with each of three thresholds (page I-269, section 2).

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the coding system of Kleijn et al to implement discriminating a voiced/unvoiced mode

is based on a past quantized gain, as taught by Ozawa et al, for the purpose of improving coding efficiency and reducing codebook size, as suggested by Ozawa et al.

Kleijn et al further teach, sound source quantization which has a codebook for representing a signal by combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discriminating section, and outputs a code vector that minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule at col. 6, lines 21-61.

Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction. However, implementation of a multiplexer and providing an analogous decoding scheme for a specific coding system was well known.

Ozawa implements a M-LCELP encoder and decoder structure, which includes multiplexer on the encoder and demultiplexer with the decoder.

Therefore, it would have been obvious to one of ordinary skill at the time of invention to implement a multiplexer and a decoding scheme with the system of Kleijn et al for the purpose of providing high quality speech coding and decoding as suggested by Ozawa et al.

Regarding claims 3, 8 and 11, Kleijn discloses a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter at col. 5, lines 66-67 and col. 6, lines 1-3;

Additionally, Kleijn discloses an adaptive codebook section for obtaining a delay and a gain and obtaining a residual by predicting a speech signal at col. 6, lines 52-62;

Kleijn discloses a discrimination section for discriminating a voiced/unvoiced mode at col. 7 lines 10-26 and col. 5, lines 7-8;

Kleijn et al do not specifically teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook. However, determination of a voiced/unvoiced mode based on a past quantized gain of an adaptive codebook was well known in the art.

In a similar field of endeavor, Ozawa et al teaches a speech coding system (implementing encoder and decoder structures) which implements mode selection such that coding methods and codebooks are changed to improve coding efficiency as well as to reduce codebook size, in accord with each of 4 modes, for unvoiced and transition segments, and voiced segments. The mode is determined by comparing the average of the pitch prediction gain with each of three thresholds (page I-269, section 2).

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the coding system of Kleijn et al to implement discriminating a voiced/unvoiced mode is based on a past quantized gain, as taught by Ozawa et al, for the purpose of improving coding efficiency and reducing codebook size, as suggested by Ozawa et al.

Kleijn et al further teach, sound source quantization which has a codebook for representing a signal by combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from the discriminating section, and a gain codebook for quantizing gains, and searches combinations of code vectors stored in said codebook, a plurality of shift amounts used to shift positions of the pulses, and gain code vectors stored in said gain codebook so as to output a combination of a code vector, shift amount, and gain code vector which minimizes distortion relative to input speech at col. 6, lines 21-61.

Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction. However, implementation of a multiplexer and providing an analogous decoding scheme for a specific coding system was well known.

Ozawa implements a M-LCELP encoder and decoder structure, which includes multiplexer on the encoder and demultiplexer with the decoder.

Therefore, it would have been obvious to one of ordinary skill at the time of invention to implement a multiplexer and a decoding scheme with the system of Kleijn et al for the purpose of providing high quality speech coding and decoding as suggested by Ozawa et al.

Regarding claim 4, Kleijn teaches a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter at col. 5, lines 66-67 and col. 6, lines 1-3;

Additionally, Kleijn teaches an adaptive codebook section for obtaining a delay and a gain and obtaining a residual by predicting a speech signal at col. 6, lines 52-62;
Discrimination section for discriminating a voiced/unvoiced mode at col. 7 lines 10-26 and col. 5, lines 7-8;

Kleijn et al do not specifically teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook. However, determination of a voiced/unvoiced mode based on a past quantized gain of an adaptive codebook was well known in the art.

In a similar field of endeavor, Ozawa et al teaches a speech coding system (implementing encoder and decoder structures) which implements mode selection such that coding methods and codebooks are changed to improve coding efficiency as well as to reduce codebook size, in

accord with each of 4 modes, for unvoiced and transition segments, and voiced segments. The mode is determined by comparing the average of the pitch prediction gain with each of three thresholds (page I-269, section 2).

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the coding system of Kleijn et al to implement discriminating a voiced/unvoiced mode is based on a past quantized gain, as taught by Ozawa et al, for the purpose of improving coding efficiency and reducing codebook size, as suggested by Ozawa et al.

Kleijn et al further teaches, sound source quantization which has a codebook for representing a signal by combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section, and a gain codebook for quantizing gains, and searches combinations of code vectors stored in said codebook, a plurality of shift amounts used to shift positions of the pulses, and gain code vectors stored in said gain codebook so as to output a combination of a code vector, shift amount, and gain code vector which minimizes distortion relative to input speech at col. 6, lines 21-61.

Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction. However, implementation of a multiplexer and providing an analogous decoding scheme for a specific coding system was well known.

Ozawa implements a M-LCELP encoder and decoder structure, which includes multiplexer on the encoder and demultiplexer with the decoder.

Therefore, it would have been obvious to one of ordinary skill at the time of invention to implement a multiplexer and a decoding scheme with the system of Kleijn et al for the purpose of providing high quality speech coding and decoding as suggested by Ozawa et al.

Regarding claims 9-10, Kleijn and Ozawa et al teach everything as claimed in claim 8. Additionally, at col.7, lines 2-26, Kleijn discloses determination of the time shift amount is based on a value that minimizes a certain criteria, which reads on "sound source quantization uses a position generated according to a predetermined rule as a pulse position when mode discrimination indicates a predetermined mode."

Response to Arguments

3. Applicant's arguments with respect to claims 1-11 have been considered but are moot in view of the new ground(s) of rejection.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Angela A. Armstrong whose telephone number is 703-308-6258. The examiner can normally be reached on Monday-Thursday 7:30-5:00 PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on (703) 305-9645. The fax phone numbers for the organization where this application or proceeding is assigned are 703-872-9314 for regular communications and 703-872-9314 for After Final communications.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the receptionist whose telephone number is 703-306-0377.

Angela A. Armstrong
Examiner
Art Unit 2654

AAA
August 11, 2003

Vijay Bhawan
8/11/03